

AUDIO SIGNAL PROCESSING CIRCUIT

Cross-Reference to Related Applications

5 The disclosure of Japanese Patent Application Nos. Hei 10-217929 and Hei 10-218128 both filed on July 31, 1998 including specification, claims, drawings and summary is herein incorporated by reference in its entirety.

10 **BACKGROUND OF THE INVENTION**

1. FIELD OF THE INVENTION

15 The present invention relates to an audio signal processing circuit in a so-called surround system. More particularly, the present invention relates to simplification of its structure, improvement of accuracy, and localization of sound image.

20 2. DESCRIPTION OF THE RELATED ART

Recently, an audio reproduction apparatus having surround channels at a left and a right sides to a listener in addition to a left and a right (and optionally a center) 25 front channels, has been developed not only for business use but also for home use. In the surround reproduction utilizing

- such apparatus, two of surround speakers are usually arranged at the both sides (i.e., left and right sides) to the listener. When the correlation between the left and the right surround signals is small (i.e., when a stereophonic surround system 5 is employed), the listener does not have an unnatural feeling. In contrast, when the correlation between the left and the right surround signals is large (i.e., when a monophonic surround system is employed), the following problem is recognized depending on the listener's position.
- 10 Specifically, when the listener is positioned at the center between the left and the right surround speakers, the listener has an unnatural feeling as if sound image was localized in the head of the listener.
- 15 In order to solve the above-mentioned problem, a technique alternatively dividing a monophonic signal into two channels with respect to each frequency component of predetermined width by using a comb type filter so as to virtually reproduce stereophonic sound, a technique 20 performing a pitch shift processing so as to reduce the correlation (e.g., THX system), and a technique performing a 90 degrees phase shift processing so as to make the correlation zero, have been proposed.
- 25 However, the above-mentioned techniques have the following problems, respectively.

According to the technique using the comb type filter so as to virtually reproduce stereophonic sound, unnaturally large sound is often reproduced when a musical instrument is used as sound source. Furthermore, the virtual stereophonic sound reproduction compromises the sound quality when the surround signals are stereophonic. Therefore, it is necessary to prevent the stereophonic sound reproduction in such a case. As a result, a change of a processing mode is required depending upon whether the surround signals are monophonic or stereophonic, which makes the overall processing complicated.

According to the technique performing the pitch shift processing such as THX system, there has been a tradeoff problem that the large amount of the pitch shift is required for reducing the correlation and that the large amount of the pitch shift lowers the sound quality. Furthermore, similar to the virtual stereophonic sound reproduction, a change of a processing mode is required depending upon whether the surround signals are monophonic or stereophonic, which makes the overall processing complicated.

The technique performing the 90 degrees phase shift processing is superior to the above-described techniques in view of the fact that the sound quality is not lowered in the

case of the stereophonic surround signals and that a change
of a processing mode is not required. However, sound image
is apt to be localized in the direction of the channel whose
phase relatively progresses, which provides the listener with
5 an unnatural feeling. This problem is especially remarkable
in the case where the left and the right surround sound sources
are virtual sound sources.

As described above, an apparatus and a method, which are
10 capable of performing the same processing independent of
whether the surround signals are monophonic or stereophonic,
preventing sound image localization in the head of the
listener so as to create sound field just as enveloping the
listener, and performing a processing which does not
15 compromise the sound quality even when the surround signals
are stereophonic, are eagerly demanded.

By the way, an audio signal processing circuit disclosed
in Japanese Laid-open Publication No. Hei 8-265899
20 (265899/1996) is shown in Figure 29. The circuit is used for
making a listener 102 to feel that sound image reproduced by
virtual speakers XL and XR is virtually localized at rear
sides to the listener 102. By utilizing the circuit, the
listener is able to feel that he/she is surrounded by the sound
25 reproduced with the speakers 104L and 104R as well as
surrounded by the sound reproduced with the virtual speakers

XL and **XR** even when the speakers **104L** and **104R** are actually arranged only in front of the listener **102**.

In the apparatus shown in Figure 29, a total of four filters **106a**, **106b**, **106c** and **106d** are used for performing the above-mentioned sound image localization. Transfer functions **H11**, **H12**, **H21** and **H22** of the respective filters are represented by the following equations:

10 $H11 = (h_{RR}h_{L'L} - h_{RL}h_{L'R}) / (h_{LL}h_{RR} - h_{LR}h_{RL})$
 $H12 = (h_{LL}h_{L'R} - h_{LR}h_{L'L}) / (h_{LL}h_{RR} - h_{LR}h_{RL})$
 $H21 = (h_{RR}h_{R'L} - h_{RL}h_{R'R}) / (h_{LL}h_{RR} - h_{LR}h_{RL})$
 $H22 = (h_{LL}h_{R'R} - h_{LR}h_{R'L}) / (h_{LL}h_{RR} - h_{LR}h_{RL})$

15 Here, h_{LL} is a transfer function from the speaker **104L** to the left ear **102L** of the listener **102**, h_{LR} is a transfer function from the speaker **104L** to the right ear **102R** of the listener **102**, h_{RL} is a transfer function from the speaker **104R** to the left ear **102L** of the listener **102**, and h_{RR} is a transfer
20 function from the speaker **104R** to the right ear **102R** of the listener **102**.

Equations $h_{LL}=h_{RR}$, $h_{LR}=h_{RL}$, $h_{L'L}=h_{R'R}$ and $h_{L'R}=h_{R'L}$ are satisfied in the equations stated above when the speakers **104L** and **104R** and the virtual speakers **XL** and **XR** are symmetrically arranged with respect to a central axis **108** through the

listener 102. As a result, equations $H_{11}=H_{22}$ and $H_{12}=H_{21}$ can be derived, so that the circuit can be obtained by utilizing total of two filters as shown in Figure 30 (such structure is referred to as "shuffler type filter"). Here, transfer functions H_{SUM} of the filters 110a and H_{DIF} of the filters 110b are represented by the following equations:

$$H_{\text{SUM}} = (ha' + hb') / 2(ha + hb)$$

$$H_{\text{DIF}} = (ha' - hb') / 2(ha - hb)$$

wherein equations $ha=h_{LL}=h_{RR}$, $hb=h_{LR}=h_{RL}$, $ha'=h_{L'L}=h_{R'R}$ and $hb'=h_{L'R}=h_{R'L}$ are satisfied.

As described above, in the case where the speakers are symmetrically arranged, sound image can be localized at the virtual speaker positions with the simple circuit.

Furthermore, a method for localizing sound image by utilizing a cross-feed filter 112 and a cross-talk cancel filter 114 as shown in Figure 31, has been proposed. The cross-talk cancel filter 114 functions to cancel cross-talk from the right speaker 104R to the left ear 102L of the listener and that from the left speaker 104L to the right ear 102R of the listener. Accordingly, the cross-talk cancel filter 114 makes it possible that a left channel signal L reaches only the left ear 102L and a right channel signal R reaches only the right ear 102R. As a result, sound image

can be localized at the desired position by adjusting the amount of the cross-talk with the cross-talk cancel filter 114.

5 The above-mentioned cross-talk cancel filter 114 can also be obtained by utilizing the shuffler type filter as shown in Figure 30. In this case, transfer functions H_{SUM} of the filters 110a and H_{DIF} of the filters 110b are represented by the following equations:

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$$H_{SUM}=ha/(2(ha+hb))$$

$$H_{DIF}=ha/(2(ha-hb)).$$

15 According to the shuffler type filter, a circuit having satisfactory sound image localization ability or satisfactory cross-talk cancel ability can be obtained only when the filters 110a and 110b are highly accurate. However, in order to make the filters accurate, the structure thereof becomes complicated. As a result, when a digital signal processor (DSP) is employed for the filters, it takes much time to perform a sound image localization processing or a cross-talk cancel processing. In contrast, when the structure of the filters is simple, the ability of the filters is insufficient.

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As described above, a shuffler type filter having a simple structure and a high accuracy is eagerly demanded for a surround system.

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SUMMARY OF THE INVENTION

An audio signal processing circuit according to the present invention is used for an audio reproduction apparatus at least having sound source located substantially at left and right sides to a listener. The audio signal processing circuit includes a phase difference control portion. The phase difference control portion receives a left channel signal for the left sound source and a right channel signal for the right sound source, controls a phase difference between the left and right channel signals so as to produce a relative phase difference in the range of 140 degrees to 160 degrees, and outputs the phase difference controlled left and right channel signals for the left and right sound source, respectively.

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The phase difference of 60 degrees causes the problem that sound image is localized in the direction of the channel whose phase relatively progresses, as in the case of the 90 degrees phase shift processing. The phase difference of 180 degrees (i.e., inverse phase) causes a listener unpleasant feeling as if the ear of the listener is pressurized, which

problem is unique to the inverse phase. In contrast, the phase difference of 140 to 160 degrees does not cause an unpleasant feeling unique to the inverse phase or produces sound image localization in the certain direction. As a result, the
5 present invention can prevent sound image of the monophonic signal from localizing in the head of the listener so as to create sound field just as enveloping the listener.

Furthermore, since only the phase difference control
10 operation is additionally performed according to the present invention, the audio reproduction according to the present invention does not compromise the sound quality even when the stereophonic signal is employed. As a result, according to the present invention, the same processing can be performed
15 independent of whether the input signal is monophonic or stereophonic.

In one embodiment of the invention, the phase difference control portion produces the relative phase difference of 140
20 degrees to 160 degrees in a frequency region ranging from 200 Hz to 1 kHz. Accordingly, the phase difference control can be effectively performed while the structure of the phase difference control portion is made simple.

25 According to another aspect of the present invention, a surround audio reproduction apparatus having a left and a

right channels in front of a listener and a left and a right surround channels at left and right sides with respect to the listener, is provided. The apparatus includes a phase difference control portion. The phase difference control 5 portion receives a left surround channel signal and a right surround channel signal, controls a phase difference between the left and the right surround channel signals so as to produce a relative phase difference in the range of 140 degrees to 160 degrees, and outputs the phase difference 10 controlled surround left and right channel signals for a left and a right surround sound source, respectively.

Accordingly, an audio reproduction apparatus capable of performing the same processing independent of whether the input signals are monophonic or stereophonic, preventing 15 sound image localization in the head of the listener so as to create sound field just as enveloping the listener, and performing a processing which does not compromise the sound quality even when the surround signals are stereophonic, can be obtained.

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In one embodiment of the invention, the left and the right surround sound sources are a virtual sound source produced by a sound image localization processing.

25 In another embodiment of the invention, the phase difference control portion produces the relative phase

difference of 140 degrees to 160 degrees in a frequency region ranging from 200 Hz to 1 kHz. Accordingly, the phase difference control can be effectively performed while the structure of the phase difference control portion is made simple.

According to another aspect of the present invention, an audio reproduction method at least utilizing sound source located substantially at left and right sides to a listener, is provided. The method includes the steps of: controlling a phase difference between a left channel signal for the left sound source and a right channel signal for the right sound source so as to produce a relative phase difference in the range of 140 degrees to 160 degrees; and outputting the phase difference controlled left and right channel signals for the left and right sound source, respectively.

According to still another aspect of the present invention, a shuffler type audio signal processing circuit is provided. The shuffler type audio signal processing circuit includes a first filter for producing a sum signal of a left channel signal and a right channel signal; and a second filter for producing a differential signal of the left channel signal and the right channel signal. In a shuffler type audio signal processing circuit, a gain of the second filter is higher than that of the first filter in a low

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frequency region. Accordingly, by making an accuracy of the second filter higher than that of the first filter in a low frequency region, the structure of the circuit can be simplified while a reduction of accuracy is prevented.

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According to still another aspect of the present invention, a shuffler type audio signal processing circuit is provided. The shuffler type audio signal processing circuit includes a first filter for producing a sum signal of a left channel signal and a right channel signal; and a second filter for producing a differential signal of the left channel signal and the right channel signal, wherein the first filter and the second filter are FIR filter, and the tap number of the second filter is larger than that of the first filter. 10 Accordingly, the structure of the circuit can be simplified while a reduction of accuracy is prevented. 15

In one embodiment of the invention, the second filter is composed of a filter bank. Accordingly, a processing margin can be increased by performing down-sampling. 20

In another embodiment of the invention, the filter bank performs down-sampling by the larger number for the lower frequency component. Accordingly, an accuracy of the second filter is made higher than that of the first filter in a low 25

frequency region, so that the structure of the circuit can be simplified while a reduction of accuracy is prevented.

According to still another aspect of the present invention, a shuffler type audio signal processing circuit is provided. The shuffler type audio signal processing circuit includes a first filter for producing a sum signal of a left channel signal and a right channel signal; and a second filter for producing a differential signal of the left channel signal and the right channel signal, wherein the first filter is FIR filter and the second filter is composed of a parallel connection of FIR filter and secondary IIR filter. Accordingly, an accuracy of the second filter is made higher than that of the first filter in a low frequency region, so that the structure of the circuit can be simplified while a reduction of accuracy is prevented. Furthermore, since a low frequency component can be processed with the secondary IIR filter, unnecessary increase of the tap number of the FIR filter can be prevented.

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In one embodiment of the invention, the second filter includes: FIR filter, and secondary IIR filter connected in parallel to the FIR filter at one of the intermediate taps or the end tap thereof. Accordingly, an accuracy of the second filter is made higher than that of the first filter in a low frequency region, so that the structure of the circuit can

be simplified while a reduction of accuracy is prevented. Furthermore, by varying an intermediate tap connected to the secondary IIR filter, optimum properties for the filter can be obtained.

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In one embodiment of the invention, the circuit is used as a cross-talk cancel filter.

10 In one embodiment of the invention, the circuit is used as a sound image localization processing filter.

15 According to still another aspect of the present invention, a filter is provided. The filter includes: FIR filter having a plurality of taps, IIR filter whose input is connected to one of the intermediate taps or the end tap of the FIR filter, and an adding means which adds outputs of the FIR filter and the IIR filter. Accordingly, a filter having desired properties can be obtained.

20 According to still another aspect of the present invention, a shuffler type audio signal processing method is provided. The method includes the steps of: performing a first filtering process for a sum signal of a left channel signal and a right channel signal; and performing a second filtering process for a differential signal of the left channel signal and the right channel signal, wherein an

accuracy of the second filtering process is higher than that of the first filtering process.

Thus, the invention described herein makes the possible
5 the advantages of: (1) providing a processing capable of performing the same processing independent of whether the input signals are monophonic or stereophonic, preventing sound image localization in the head of the listener so as to create sound field just as enveloping the listener, and
10 performing a processing which does not compromise the sound quality even when the surround signals are stereophonic; and (2) providing a shuffler type filter having a simple structure and a high accuracy.

15 These and other advantages of the present invention will become apparent to those skilled in the art upon reading and understanding the following detailed description with reference to the accompanying figures.

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BRIEF DESCRIPTION OF THE DRAWINGS

Figure 1 is a block diagram of an audio signal processing circuit according to an embodiment of the present invention.

Figure 2 is a block diagram of an audio reproduction apparatus wherein the audio signal processing circuit of Figure 1 is incorporated.

5 Figures 3A and 3B are circuit diagrams according to embodiments wherein an all pass filter used in the present invention is composed of an analog circuit.

10 Figure 4 is a graph illustrating a frequency-phase relationship of the all pass filter used in the present invention.

15 Figure 5 is a schematic view illustrating an arrangement of speakers in accordance with a surround audio reproduction apparatus of the present invention.

20 Figure 6 is a block diagram according to an embodiment wherein the audio signal processing circuit of the present invention is applied to a surround audio reproduction apparatus which produces virtual sound sources by a sound image localization processing using DSP.

Figure 7 is a schematic view illustrating an example of an arrangement of the virtual sound sources of Figure 6.

Figure 8 is a signal-flow diagram illustrating the sound image localization processing using DSP.

5 Figure 9 is a signal-flow diagram illustrating an embodiment wherein an all pass filter used in the present invention is composed of a secondary IIR filter.

Figure 10 is a signal-flow diagram according to another embodiment of the present invention.

10 Figure 11 is a schematic view illustrating an example of an arrangement of the virtual sound sources of Figure 10.

15 Figure 12 is a schematic view of a shuffler type filter according to an embodiment of the present invention.

Figure 13 is a block diagram illustrating a hardware structure of the audio reproduction apparatus using DSP.

20 Figure 14 is a signal-flow diagram illustrating processings carried out by the DSP in accordance with program(s) stored in a memory.

25 Figure 15 is a graph illustrating a frequency response H_{SUM} of a first filter and a frequency response H_{DIF} of a second filter, and a cross-talk cancel response $Zt1$ and a cross-

talk cancel error Z_{t2} when the first and the second filters are used, wherein both of the first and the second filters have 32 taps.

5 Figure 16 is a graph illustrating H_{SUM} , H_{DIF} , Z_{t1} and Z_{t2} wherein both of the first and the second filters have 64 taps.

Figure 17 is a graph illustrating H_{SUM} , H_{DIF} , Z_{t1} and Z_{t2} wherein both of the first and the second filters have 96 taps.

10 Figure 18 is a graph illustrating H_{SUM} , H_{DIF} , Z_{t1} and Z_{t2} wherein the first filter has 32 taps and the second filter has 96 taps.

15 Figure 19 is a signal-flow diagram according to an embodiment using a filter bank.

Figure 20 is a graph illustrating a cross-talk cancel response Z_{t1} and a cross-talk cancel error Z_{t2} when the 20 cross-talk cancel filter shown in Figure 14 is used wherein a first filter having 32 taps and a second filter having 128 taps are incorporated.

Figure 21 is a graph illustrating a cross-talk cancel 25 response Z_{t1} and a cross-talk cancel error Z_{t2} when the cross-talk cancel filter shown in Figure 19 is used wherein

a first filter having 32 taps and a second filter corresponding to 128 taps are incorporated.

Figure 22 is a signal-flow diagram according to an embodiment wherein the second filter 120b is composed of a parallel connection of FIR filter and IIR filter.

Figure 23 is a graph illustrating a frequency response H_{SUM} of the first filter and a frequency response H_{DIF} of the second filter, and a cross-talk cancel response $Zt1$ and a cross-talk cancel error $Zt2$ when the cross-talk cancel filter shown in Figure 22 is used.

Figure 24 is a signal-flow diagram according to an embodiment wherein an intermediate tap of FIR filter is connected to an input of IIR filter.

Figure 25 is a graph illustrating a desired impulse response for the second filter.

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Figure 26 is a graph illustrating an impulse response of IIR filter having properties approximate to that of Figure 25.

Figure 27 is a graph illustrating a deviation of the impulse response of the IIR filter from the desired impulse response.

5 Figure 28 is a graph illustrating an impulse response of FIR filter obtained in due consideration of the deviation of Figure 27.

10 Figure 29 is a schematic view illustrating conventional sound image localization technique.

Figure 30 is a circuit diagram illustrating shuffler type filter.

15 Figure 31 is a block diagram of a sound image localization circuit including a cross-feed filter and a cross-talk cancel filter.

DETAILED DESCRIPTION OF THE PREFERRED EMBODIMENTS

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Figure 1 is a block diagram of an audio signal processing circuit according to an embodiment of the present invention. The audio signal processing circuit includes a phase difference control portion 2. The phase difference control portion 2 receives a left channel signal s_L for a left sound source s_{SL} located substantially at a left side to a listener

(shown in Figure 5) and a right channel signal s_R for a right sound source s_{SR} located substantially at a right side to the listener (also shown in Figure 5). The phase difference control portion 2 controls a phase difference between the left 5 and right channel signals s_L and s_R so that the relative phase difference be from 140 degrees to 160 degrees (and preferably about 150 degrees) and outputs the phase difference controlled signals s'_L and s'_R for the left and right sound source, respectively.

10

The signals s'_L and s'_R processed in the above-mentioned manner are respectively supplied to the sound sources s_{SL} and s_{SR} . As a result, with respect to a monophonic signal, the circuit is capable of preventing sound image localization in 15 the head of the listener and creating sound field just as enveloping the listener. Furthermore, with respect to a stereophonic signal, the circuit is capable of performing a processing which does not compromise the sound quality (i.e., a feeling that sound image of the left and the right surround 20 channels is comfortably localized).

Figure 2 is a block diagram of an audio signal processing circuit 4 which is incorporated into an audio reproduction apparatus, wherein the phase difference control portion 2 includes all pass filters (APFs) 6 and 8. The apparatus 25 includes an amplifier and speakers both of which are connected

to the output of the audio signal processing circuit 4 (not shown in Figure 2).

A central channel signal **C**, a front left channel signal
5 **F_L**, a front right channel signal **F_R**, a surround left channel
signal **S_L**, a surround right channel signal **S_R**, and a low
frequency channel signal **LFE** are input to the circuit 4. Among
these signals, The central channel signal **C**, the front left
channel signal **F_L**, the front right channel signal **F_R**, and the
10 low frequency channel signal **LFE** are output without any
processing. The surround left channel signal **S_L** is processed
with the APF 6 so as to be output as the signal **S'_L**. The
surround right channel signal **S_R** is processed with the APF 8
so as to be output as the signal **S'_R**. In this embodiment, the
15 APFs 6 and 8 constitute the phase difference control portion
2.

An example of the APF 6 is shown in Figure 3A. The example
illustrates secondary APF. A frequency-phase relationship
20 of the APF 6 is shown as a curved line 10 in Figure 4. In
a low frequency region, the phase of the output signal is the
same as that of the input signal (i.e., the phase difference
between the input and the output signals is zero). The phase
of the output signal delays as the frequency increases, and
25 in a high frequency region, the phase of the output signal
becomes again the same as that of the input signal (i.e., the

phase difference between the input and the output signals becomes 360 degrees). In other words, the phase difference between the input and the output signals varies in the range of zero to 360 degrees depending upon the frequency. The 5 properties of the APF 6 represented by the curved line 10 may be adapted by selecting resistance **R1** and **R2** and capacitor **C1** and **C2**.

A desired phase difference $\arg(S'_R/S'_L)$ is represented by 10 the following equation:

$$\arg(S'_R/S'_L) = \arg(S'_R/S_R) - \arg(S'_L/S_L)$$

here, the following equations are satisfied:

15

$$\arg(S'_L/S_L) = \tan^{-1}((-2(f/f_1))/(1-(f/f_1)^2)) + \tan^{-1}((-2(f/f_2))/(1-(f/f_2)^2))$$

$$\arg(S'_R/S_R) = \tan^{-1}((-2(f/f_3))/(1-(f/f_3)^2)) + \tan^{-1}((-2(f/f_4))/(1-(f/f_4)^2))$$

20 $f_1 = 1/(2\pi C_1 R_1)$

$$f_2 = 1/(2\pi C_2 R_2)$$

$$f_3 = 1/(2\pi C_3 R_3)$$

$$f_4 = 1/(2\pi C_4 R_4).$$

25 Therefore, the APF 6 having desired properties can be designed based on the above-mentioned equations.

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An example of the APF 8 is shown in Figure 3B. The structure thereof is basically the same as that of the APF 6. The properties of the APF 8 represented by a curved line 5 12 of Figure 4 are obtained by selecting resistance R3 and R4 and capacitor C3 and C4. By utilizing the above-mentioned APFs 6 and 8, the phase difference of 140 to 160 degrees can be obtained between the surround left channel signal S'_L and the surround right channel signal S'_R in a frequency region 10 ranging from 200 Hz to 1 kHz. In other words, when the monophonic surround left channel signal S_L and the monophonic surround right channel signal S_R are supplied to the APFs 6 and 8, the APFs 6 and 8 can control the phase difference between the signals S_L and S_R so that the phase of the signal 15 S'_R relatively progresses or delays 140 to 160 degrees to that of the signal S'_L .

The output signals obtained in the above-mentioned manner are supplied to respective speakers as shown in Figure 20 5. More specifically, the central channel signal C is supplied to a speaker S_c ; the front left channel signal F_L is supplied to a speaker S_{FL} ; the front right channel signal F_R is supplied to a speaker S_{FR} ; and the low frequency channel signal LFE is supplied to a speaker S_{LFE} . Furthermore, the 25 surround left channel signal S'_L is supplied to a speaker S_{SL} ,

and the surround right channel signal s'_R is supplied to a speaker s_{SR} .

Alternatively, the relative phase difference of 140 to
5 160 degrees can be obtained by producing a phase difference
of 20 to 40 degrees between the channels with APFs and then
inversing the phase of one of the channels.

Although the desired phase difference is produced in the
10 frequency region of 200 Hz to 1 kHz according to the
above-mentioned embodiment, it is more preferred if the
desired phase difference can be obtained in the frequency
region of 50 Hz to 4 kHz. The higher order of the APFs widens
the frequency band wherein the desired phase difference is
15 obtained.

Although the above-mentioned embodiment has illustrated
the case where the surround speakers s_{SL} and s_{SR} are arranged
at just the left and the right sides to the listener 50, the
20 surround speakers s_{SL} and s_{SR} may be arranged in an angular
range represented by α of Figure 5. In Figure 5, the angle
range α of 60 degrees (more specifically, 30 degrees both in
front and in rear with respect to the line connecting the
surround speakers s_{SL} and s_{SR}) is exemplified. Accordingly,
25 in the present specification, the phrase "substantially at

left and right sides to a listener" is meant to be the above-mentioned angular range α .

Figure 6 shows a surround audio reproduction apparatus creating virtual sound sources with DSP, wherein the phase difference control portion in accordance with the present invention is incorporated. The respective input signals **C**, **F_L**, **F_R**, **S_L**, **S_R** and **LFE** are obtained by decoding a digitized data converted from an analog signal with an A/D converter or a digital-bit-stream encoded for surround, with a multi-channel surround decoder (not shown). The respective input signals are supplied to the DSP 22. The multi-channel surround decoder can either be incorporated into the DSP or separately provided therefrom.

A signal for a left speaker **L_{OUT}**, a signal for a right speaker **R_{OUT}** and a signal for a sub-woofer speaker **SUB_{OUT}** are produced by performing processings such as addition, subtraction, filtering, delay and the like with the DSP 22 to the thus-input digital data in accordance with program(s) stored in a memory 26. The thus-produced signals are converted into analog signals with a D/A converter 24 and are supplied to the speakers **S_{FL}**, **S_{FR}** and **S_{LFE}**. Installation process of the program(s) into the memory 26 and other processings are carried out by a micro-processor 20.

In this embodiment, it is presumed that the speakers S_{FL} and S_{FR} and the virtual surround sound sources X_{SL} and X_{SR} are symmetrically arranged with respect to the central axis 40 through the listener as shown in Figure 7. Since bass (sound having a low frequency) reproduced by the woofer speaker S_{LFE} has a weak directivity and a long wavelength, the woofer speaker S_{LFE} can be arranged at any location.

Figure 8 is a signal-flow diagram illustrating processings carried out by the DSP 22 in accordance with the program(s) stored in the memory 26. According to this embodiment, as shown in Figure 7, the virtual central sound source X_c , the virtual surround left sound source X_{SL} and the virtual surround right sound source X_{SR} are created by using only the front left and right speakers S_{FL} and S_{FR} and the low frequency speaker S_{LFE} .

The surround left channel signal S_L and the surround right channel signal S_R are subjected to a sound image localization processing with a surround sound image localization circuit 12 and are supplied to the left and the right speakers S_{FL} and S_{FR} arranged in front of the listener. The surround sound image localization circuit 12 is composed of a so-called shuffler type filter. Therefore, the effect that the surround left channel signal S_L and the surround right channel signal S_R are output respectively from the virtual

surround left sound source \mathbf{x}_{SL} and the virtual surround right sound source \mathbf{x}_{SR} can be obtained.

The central channel signal \mathbf{C} is equally supplied to the
5 left and the right speakers \mathbf{S}_{FL} and \mathbf{S}_{FR} . Therefore, the effect
that the central channel signal \mathbf{C} is output from the virtual
central sound source \mathbf{x}_c can be obtained.

Delay processing circuits **14L**, **14R** and **30** provide a delay
10 time equal to that caused by the surround sound image
localization circuit **12**. These delay circuits can compensate
the delay between the signals \mathbf{C} , \mathbf{F}_L , \mathbf{F}_R and **LFE** and the signals
 \mathbf{S}_L and \mathbf{S}_R .

15 The surround left channel signal \mathbf{s}_L and the surround
right channel signal \mathbf{s}_R are subjected to a phase difference
control processing with the phase difference control portion
2 in the above-mentioned manner before being supplied to the
surround sound image localization circuit **12**. Therefore, a
20 relative phase difference of 140 to 160 degrees has already
been produced between the surround left channel signal \mathbf{s}_L and
the surround right channel signal \mathbf{s}_R .

In this embodiment, a secondary IIR filter as shown in
25 Figure **9** is used as the APFs **6** and **8** constituting the phase
difference control portion **2**.

Since the phase difference control processing is performed with the phase difference control portion 2, the surround left channel signal S_L output from the virtual surround left sound source X_{SL} and the surround right channel signal S_R output from the virtual surround right sound source X_{SR} may be prevented from being localized in the head of the listener 50.

Figure 10 is a signal-flow diagram according to another embodiment of the present invention. According to this embodiment, the front left channel signal F_L and the front right channel signal F_R are respectively added to the surround left channel signal S_L and the surround right channel signal S_R which have already been subjected to the phase difference control processing. As a result, as shown in Figure 11, the front left channel signal F_L is localized at the position of the virtual sound source X_{FL} located between the positions of the left speaker S_{FL} and the virtual surround left sound source X_{SL} . Likewise, the front right channel signal F_R is localized at the position of the virtual sound source X_{FR} located between the positions of the right speaker S_{FR} and the virtual surround right sound source X_{SR} . Accordingly, sound field created by the front left channel signal F_L and the front right channel signal F_R can be widen.

In the above embodiments, an analog circuit can be used in place of the described digital circuit and a digital circuit can be used in place of the described analog circuit.

5 Figure 12 is a schematic view of a shuffler type cross-talk cancel filter 130 according to an embodiment of the present invention. A left channel signal is supplied to a left channel input terminal L_{IN} and a right channel signal is supplied to a right channel input terminal R_{IN} . The left
10 and the right channel signals are added up with an adder 122 and the added signal is supplied to a first filter 120a. The right channel signal is subtracted from the left channel signal with a subtracter 124 and the subtracted signal is supplied to a second filter 120b. Transfer functions H_{SUM} and
15 H_{DIF} of the first and the second filters 120a and 120b are represented by the following equations, respectively:

$$H_{SUM} = ha/2 (ha+hb)$$

$$H_{DIF} = ha/2 (ha-hb)$$

20 An adder 126 adds the outputs of the first and the second filters 120a and 120b and outputs a signal for a speaker 104L. A subtracter 128 subtracts the outputs of the second filter 120b from the output of the first filter 120a and outputs a
25 signal for a speaker 104R.

According to this embodiment, the first and the second filters **120a** and **120b** are FIR filters and the cross-talk cancel filter **130** is composed of DSP. Figure **13** is a block diagram illustrating a hardware structure of the audio reproduction apparatus using DSP **140**. A left and a right channel signals **L** and **R** are supplied as digital data to the DSP **140**. A signal for a left speaker **L_{OUT}** and a signal for a right speaker **R_{OUT}** are produced by performing processings such as addition, subtraction, filtering, delay and the like with the DSP **140** to the thus-input digital data in accordance with program(s) stored in a memory **146**. The thus-produced signals are converted into analog signals with a D/A converter **142** and are supplied to the speakers **104L** and **104R**.

Installation process of the program(s) into the memory **26** and other processings are carried out by a micro-processor **120**.

Figure **14** is a signal-flow diagram illustrating processings carried out by the DSP **140** in accordance with the program(s) stored in the memory **146**. According to this embodiment, the first and the second filters **120a** and **120b** are FIR filters. In Figure **14**, **DS1** to **DS31** and **DD1** to **DD95** denote delay means. The delay means perform delay processing in an amount of one sampling data. In this embodiment, the sample frequency is set to be 48 kHz. **KS0** to **KS31** and **KD0** to **KD95** denote coefficient processing means. In this embodiment, the tap number (i.e., the number of the

coefficient processings) of the first filter **120a** is set to be 32 and the tap number of the second filter **120b** is set to be 96. In the case of FIR filter, the larger tap number produces the higher accuracy in a low frequency region.

5 Accordingly, in the example of Figure **14**, the accuracy of the second filter **120b** is higher than that of the first filter **120a** in a low frequency region.

Figure **15** shows a frequency response H_{SUM} of the first
10 filter **120a** and a frequency response H_{DIF} of the second filter
120b wherein the first and the second filters have 32 taps.
Figure **15** also shows a cross-talk cancel response Z_{t1} and a
cross-talk cancel error Z_{t2} when a cross-talk cancel filter
wherein the first and the second filters are incorporated is
15 used. Here, the error is meant to be a remained response
(i.e., a response that had not been sufficiently canceled).
Therefore, regarding the cross-talk cancel filter, the better
filter produces the smaller error. In this embodiment, an
angle β defined by the speaker **104L** (or **104R**) and the listener
20 **102** as shown in Figure **12** is set to be 10 degrees. As shown
in Figure **15**, when the tap number of the first and the second
filters **120a** and **120b** is 32, the accuracy is low and a large
cross-talk cancel error is caused.

25 Figure **16** shows a frequency response H_{SUM} of the first
filter **120a** and a frequency response H_{DIF} of the second filter

120b wherein the first and the second filters have 64 taps. Figure **16** also shows a cross-talk cancel response $Zt1$ and a cross-talk cancel error $Zt2$ when a cross-talk cancel filter wherein the first and the second filters are incorporated is used. Figure **16** shows that, although the cross-talk cancel properties are improved compared to the case of 32 taps shown in Figure **15**, the cross-talk cancel error is still large.

Figure **17** shows a case where the first and the second filters **120a** and **120b** have 96 taps. Figure **17** shows that the cross-talk cancel error is small. However, in this case, the problem that an arithmetical load to DSP **140** is large arises.

According to this embodiment, the tap number of the first filter **120a** is set to be smaller than that of the second filter **120b** in view of the fact that a frequency response required for the first filter **120a** is low level and flat especially in a low frequency region. In other words, the accuracy of the first filter **120a** is set to be low in a low frequency region and the accuracy of the second filter **120b** is set to be higher instead. More specifically, the tap number of the first filter **120a** is set to be 32 and the tap number of the second filter **120b** is set to be 96. Frequency response H_{SUM} and H_{DIF} , a cross-talk cancel response $zt1$ and a cross-talk cancel error $zt2$ in this case are shown in Figure **18**.

As is apparent from Figure 18, the error in this case is as small as that in the case where the tap numbers of the first and the second filters **120a** and **120b** are both 96. According to this embodiment, a shuffler type cross-talk 5 cancel filter having high accuracy can be obtained while keeping low a total tap number thereof.

Figure 19 is a signal-flow diagram according to another embodiment of the present invention. FIR filters are also 10 employed in this embodiment. Furthermore, the tap number of the second filter **120b** is set to be larger than that of the first filter **120a**. More specifically, the tap number of the second filter **120b** is set to correspond to 128 and the tap number of the first filter **120a** is set to be 32. In addition, 15 a filter bank is employed for the second filter **120b** according to this embodiment. As a result, down-sampling is performed with respect to the signal supplied to the second filter **120b** and then the signal is processed with the FIR filters. In figure 19, **H** denotes a high-pass filter, **G** denotes a low- 20 pass filter, the arrow ↓ denotes down-sampling by 2 and the arrow ↑ denotes up-sampling by 2. Delay means **205**, **206** and **208** perform delay processing which compensates a time required for the processing performed by the filter bank. The delay means **205** performs delay processing in an amount of 25 three sampling data, the delay means **206** performs delay processing in an amount of one sampling data, and the delay

means **208** performs delay processing in an amount of seven sampling data.

According to this embodiment employing the filter bank,
5 a cross-talk cancel filter having a high ability of 128 taps can be obtained while the total tap number of the FIR filters **201**, **202**, **203** and **204** is kept 68 taps. In other words, a processing margin can be increased by performing down-sampling. As a result, the accuracy in a low frequency
10 component can be improved. Although a so-called octave dividing filter bank has been exemplified in this embodiment, a so-called equal dividing filter bank may also be employed. According to the octave dividing filter bank, a frequency component is divided in a geometrical ratio preferentially
15 in a lower frequency side. In contrast, according to the equal dividing filter bank, a frequency component is equally divided with respect to an overall frequency region.

Figure **20** shows a cross-talk cancel error ZT2 in the case
20 where the tap number of the first filter **120a** is 32 and the tap number of the second filter **120b** is 128 and where a filter bank is not employed. Figure **21** shows a cross-talk cancel error ZT2 when the cross-talk cancel filter shown in Figure
19 is used. As is apparent from the comparison between Figures
25 **20** and **21**, the circuit of Figure **19** which employs a filter

bank has the ability as good as that of the circuit having actually 128 taps.

Figure 22 is a signal-flow diagram according to still 5 another embodiment of the present invention. According to this embodiment, the first filter 120a is FIR filter having 32 taps and the second filter 120b is composed of a parallel connection of FIR filter 210 having 32 taps and secondary IIR filter 212. The outputs of the FIR filter 210 and the 10 secondary IIR filter 212 are added up with an adder 214.

According to this embodiment, an accuracy with respect to a low frequency component can be improved by utilizing the secondary IIR filter 212 while the tap number of the FIR filter 15 210 in the second filter is kept 32 taps. Since the secondary IIR filter produces a higher accuracy in a low frequency region, the cross-talk cancel filter according to this embodiment produces an accuracy as high as the filter of Figure 12 wherein both of the first and the second filters 20 are FIR filters, while the tap number of the filter according to this embodiment is smaller than that of the filter of Figure 12. Although the secondary IIR filter has been exemplified in this embodiment, IIR filter of the first order or the higher order may also be employed. The IIR filter of the higher order 25 can be composed of either series connection or parallel connection.

Figure 23 shows a frequency response H_{SUM} of the first filter 120a and a frequency response H_{DIF} of the second filter 120b in the circuit (i.e., the cross-talk cancel filter) of Figure 22. Figure 23 also shows a cross-talk cancel response Zt_1 and a cross-talk cancel error Zt_2 of the circuit of Figure 22. As is apparent from Figure 23, accuracy substantially as high as that of the case shown in Figure 18 is obtained.

According to the embodiment shown in Figure 22, the second filter 120b, which is composed of parallel connection of the FIR filter and the secondary IIR filter, is exemplified. However, as shown in Figure 24, one of intermediate taps of the FIR filter can be connected to the input of the secondary IIR filter. The end tap (i.e., the tap of the number $m-1$ in Figure 24) may also be connected to the input of the secondary IIR filter. As a result, properties of the second filter 120b can be easily varied depending upon the desired properties.

Hereinafter, a design method of the filter shown in Figure 24 will be described with reference to Figures 25 to 28. Figure 25 shows an impulse response required for the second filter 120b. Based on the required impulse response, an impulse response of the secondary IIR filter is decided. Initially, the impulse response is decided by preferentially

approximating it to the latter part of the required impulse response (which corresponds to a low frequency region), as shown in Figure 26. In the example of Figure 26, the impulse response of the secondary IIR filter having the property
5 approximate to that of the required impulse response after the sample of the number k is obtained. It is noted that; with respect to the sample of the number k to the sample of the number m, the impulse response of the secondary IIR filter is largely deviated from the required impulse response.

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Next, the impulse response of the FIR filter is obtained with respect to the sample of the number zero to the sample of the number m. As described above and as shown in Figure 27, the impulse response of the secondary IIR filter is
15 largely deviated from the required impulse response with respect to the sample of the number k to the sample of the number m. In consideration of such a deviation, the impulse response of the FIR filter as shown in Figure 28 is obtained with respect to the sample of the number zero to the sample
20 of the number m.

As described above, the second filter 120b as shown in Figure 24 can be obtained. The intermediate tap connected to the input of the secondary IIR filter is the tap
25 corresponding to the first sample from which the approximation is conducted (i.e., the sample of the number

k in the above-mentioned example). As described above, a filter having a desired impulse response can be easily obtained.

5 In the above embodiments, the tap number has been described only for being exemplified. Furthermore, the cross-talk cancel filter has been described in the above embodiments, however, the present invention is applicable to a sound image localization filter.

10 In the above embodiments, FIR filter is used for the first filter **120a**. However, the first filter **120a** may also be composed of a parallel connection of FIR filter and IIR filter (as shown in Figures **22** and **24**). Alternatively, the 15 first filter **120a** may employ a filter bank. Even in this case, when the second filter **120b** having a higher accuracy than that of the first filter **120a** is employed, a cross-talk cancel filter having a high accuracy can be obtained while keeping simple an overall structure of the filter.

20 In the above embodiments, although DSP is used in the cross-talk cancel filter, an analog filter may be entirely or partially substituted for the DSP.

25 Various other modifications will be apparent to and can be readily made by those skilled in the art without departing

from the scope and spirit of this invention. Accordingly, it is not intended that the scope of the claims appended hereto be limited to the description as set forth herein, but rather that the claims be broadly construed.

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